

Application Serial No.: 10/799,505
Attorney Docket No.: 0160114

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List of Claims:

APR 11 2008

1. **(Currently Amended)** A method for suppressing background noise from a speech signal, said method comprising:
 - obtaining an input speech signal;
 - performing linear predictive coding (LPC) analysis on said input speech signal to obtain a z-domain representation of said input speech signal;
 - computing a spectrum tilt and a noise-to-signal ratio (NSR) of said z-domain representation of said input speech signal;
 - obtaining a spectrum tilt of a background noise model;
 - applying a gain to reduce energy of said input speech signal when said NSR is high;
 - reducing a spectral valley energy of said input speech signal when said spectrum tilt of said input speech signal is equivalent to said spectrum tilt of said background noise model; and
 - applying an inverse filter to said input speech signal when said spectrum tilt of said input speech signal is not equivalent to said spectrum tilt of said background noise model, wherein said inverse filter is an inverse of [[said]] a z-domain representation of said background noise model.
2. **(Previously Presented)** The method of claim 1, wherein said input speech signal comprises a plurality of sub-frames processed in sequence.
3. **(Previously Presented)** The method of claim 1, wherein said gain is adaptive based on characteristics of said input speech.

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4. (Previously Presented) The method of claim 1, wherein said background noise model is a first order model.

5. (Currently Amended) A computer program product comprising:
a computer usable medium having computer readable program code embodied therein for suppressing background noise from a speech signal; said computer readable program code configured to cause a computer to:

- obtain an input speech signal;
- perform linear predictive coding (LPC) analysis on said input speech signal to obtain a z-domain representation of said input speech signal;
- compute a spectrum tilt and a noise-to-signal ratio (NSR) of said z-domain representation of said input signal;
- obtain a spectrum tilt of a background noise model;
- apply a gain to reduce energy of said input speech signal when said NSR is high;
- reduce a spectral valley energy of said input speech signal when said spectrum tilt of said input speech signal is equivalent to said spectrum tilt of said background noise model; and
- apply an inverse filter to said input speech signal when said spectrum tilt of said input speech signal is not equivalent to said spectrum tilt of said background noise model, wherein said inverse filter is an inverse of [[said]] a z-domain representation of said background noise model.

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6. **(Previously Presented)** The computer program product of claim 5, wherein said input speech signal comprises a plurality of sub-frames processed in sequence.

7. **(Previously Presented)** The computer program product of claim 5, wherein said gain is adaptive based on characteristics of said input speech.

8. **(Previously Presented)** The computer program product of claim 5, wherein said background noise model is a first order model.

9. **(Currently Amended)** An apparatus for suppressing background noise from a speech signal, said apparatus comprising:

an object for receiving an input speech signal;

an object for performing linear predictive coding (LPC) analysis on said input speech signal to obtain a z-domain representation of said input speech signal;

an object for computing a spectrum tilt and a noise-to-signal ratio (NSR) of said z-domain representation of said input signal;

an object for obtaining a spectrum tilt of a background noise model;

an object for applying a gain to reduce energy of said input speech signal when said NSR is high;

an object for reducing a spectral valley energy of said input speech signal when said spectrum tilt of said input speech signal is equivalent to said spectrum tilt of said background noise model; and

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an object for applying an inverse filter to said input speech signal when said spectrum tilt of said input speech signal is not equivalent to said spectrum tilt of said background noise model, wherein said inverse filter is an inverse of ~~[[the]]~~ a z-domain representation of said background noise model.

10. (Previously Presented) The apparatus of claim 9, wherein said input speech signal comprises a plurality of sub-frames processed in sequence.

11. (Previously Presented) The apparatus of claim 9, wherein said gain is adaptive based on characteristics of said input speech.

12. (Previously Presented) The apparatus of claim 9, wherein said background noise model is a first order model.

13. (Previously Presented) The method of claim 1, wherein applying said gain, reducing said spectral valley energy and applying said inverse filter are performed using $g \cdot [1/F_n(z/a)] \cdot F_s(z/b)/F_s(z/c)$, wherein parameters a ($0 \leq a < 1$), b ($0 < b < 1$), and c ($0 < c < 1$) are adaptive coefficients, and parameter g is an adaptive gain.

14. (Previously Presented) The method of claim 13, wherein said parameters a , b , c , and g are controlled by said NSR.

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15. (Previously Presented) The computer program product of claim 5, wherein said computer readable program code to apply said gain, reduce said spectral valley energy and apply said inverse filter are performed using $g \cdot [1/F_n(z/a)] \cdot F_s(z/b)/F_s(z/c)$, wherein parameters a ($0 \leq a < 1$), b ($0 < b < 1$), and c ($0 < c < 1$) are adaptive coefficients, and parameter g is an adaptive gain.

16. (Previously Presented) The computer program product of claim 15, wherein said parameters a , b , c , and g are controlled by said NSR.

17. (Previously Presented) The apparatus of claim 9, wherein said objects for applying said gain, reducing said spectral valley energy and applying said inverse filter are performed using $g \cdot [1/F_n(z/a)] \cdot F_s(z/b)/F_s(z/c)$, wherein parameters a ($0 \leq a < 1$), b ($0 < b < 1$), and c ($0 < c < 1$) are adaptive coefficients, and parameter g is an adaptive gain.

18. (Previously Presented) The apparatus of claim 17, wherein said parameters a , b , c , and g are controlled by said NSR.